REMARKS

Claims 1-10 were again presented for reconsideration and reexamination and, in the aforementioned Office action, were all rejected under either 35 U.S.C.§103(a) as allegedly unpatentable over various combinations of cited references. The Examiner has withdrawn the rejections based on Morgan et al. (US 5,329,587) in view of Romesburg (US 5,796,819) and has now rejected the claims as allegedly unpatentable over Marash (US 5,825,898) in view of Coker (US 4,581,758). The Examiner has made the rejection final. By this amendment, Applicant traverses the rejection. Claims 1-10 remain in the application.

FINALITY OF REJECTION:

Applicant respectfully requests reconsideration of the finality of the rejection. Although Applicant no longer has the right to amend so long as the Examiner presents new references or reasons for rejection, Applicant notes that MPEP §706.07 provides guidelines for final rejection, and those guidelines appear to be applicable to the present case. In particular, MPEP §706.07 provides that "[b]efore final rejection is in order a clear issue should be developed between the examiner and applicant." The section also notes that "[s]witching ... from one set of references to another by the examiner in rejecting successive actions claims of substantially the same subject matter, will ... tend to defeat attaining the goal of reaching a clearly defined issue for an early termination ...." From Applicant's viewpoint, this is exactly what has happened here. The Examiner previously cited a combination of references in a Section 103 rejection of the claims, and has now withdrawn those rejections in favor of new ones based on a completely different set of references, which were not previously cited.

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Reliance on MPEP §706.07(a) to make the rejection final is, Applicant feels, misplaced in this instance. In a broad sense, a new ground of rejection is always necessitated by Applicant's amendment, but to apply this rule when the Examiner is substituting a brand new set of references seems to be totally at odds with the guidelines for final rejections set forth at the beginning of MPEP §706.07. If the finality of the rejection is maintained, Applicant will be forced to appeal the rejection prematurely, before a clear issue has been developed between the Examiner and Applicant. This is surely not what was intended by MPEP §706.07. Accordingly, Applicant respectfully requests the Examiner to reconsider and withdraw the finality of the rejection.

REJECTION UNDER 35 U.S.C. §103(a):

Claims 1-10 were rejected as allegedly unpatentable over Marash in view of Coker.

In paragraph 3 of the Office action, the Examiner characterizes Marash as teaching "a system and method for adaptive interference canceling for reducing interference in a signal received from an array of sensors." The key word in this characterization is "canceling." The abstract of Marash makes it clear that "[a]daptive filters are used to generate cancelling signals that closely approximate the interference present in the received signal." A careful reading of Marash reveals that it functions by generating an estimate of the interference (noise) components in the received signal and correcting the received signal by subtraction of the estimate (in difference unit 8). This fundamental feature of Marash is not shared by the present invention. The present invention does not compute an estimate of noise and subtract the estimate

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from the received signal, as will be further explained below. It is Applicant's view, for this reason, that Marash is simply not pertinent to the present invention as claimed.

In the Marash system, signals are received at multiple sensors. The received signals include a signal of interest from a single source and multiple interference signals. After sampling, the received signals are processed along two parallel paths, initially through a main channel matrix 3 and a reference channel matrix 4. Although the function of each of these matrices is not immediately obvious from the FIG. 1 of the drawings, they are clearly described in the specification with reference to FIGS. 5 and 6. The main channel matrix 3 is a beam former or spatial filter that filters the received signals in order to maximize sensitivity toward the signal source. (See column 6, lines 15-21.) Similarly, the reference channel matrix 4 is also a beamformer, but designed to present a null toward the direction of the signal source, thereby steering the input array to obtain interference signals from directions other than that of the signal source. (See column 6, lines 39-55.) Further processing of the interference signals, through decolorizing filters 5 and frequency selective constraint adaptive filters 6, results in a set of estimates for the interference signals, on lines 15, and these are subtracted from the input signal representative of the signal source, in difference unit 8. Marash recognized that, even though the main channel matrix 3 steers the sensor array toward the signal source, some components of the interference signals will find their way through the main channel matrix. (See column 6, lines 22-25.) Therefore, the system provides a way of canceling these components. In brief, the Marash system reduces the effect of interference signals by a combination of beam steering and interference estimation and cancellation.

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By way of contrast, the present invention in its broadest form does not use beam steering, although beam steering could be used in conjunction with the invention. Further, the present invention does not estimate a noise component and subtract the estimated noise from the received signals. In concept, the present invention is much simpler than the Marash disclosure, or any other systems that perform beam steering and interference/noise estimation. In brief, the present invention simply combines by addition the signals received at multiple microphones. One of the microphones is designated a "reference" microphone and the signals from the other microphones are aligned with the reference microphone signal, so that they can be combined by addition in a summation circuit. The significance of the summation circuit is that the microphone signal components derived from the signal source of interest will be coherent with each other, while the signal components derived from the interference or noise sources will be incoherent. Thus, the summation circuit provides an output that has an enhanced signal-to-noise ratio. In effect, the noise or interference components are reduced, although not completely eliminated, but there has been no need to resort to beam steering or to noise component estimation and subtraction. As described and illustrated in Applicant's application, this technique provides significant noise reduction, but it is achieved without the complexity and associated cost of systems such as that of Marash.

In paragraph 4 of the Office action, the Examiner sets forth the alleged similarities of the Marash system and the present invention. For completeness, the points raised by the Examiner are analyzed here:

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The Examiner first asserts that "[r]egarding claims 1 and 6, at column 4, lines 50-67 Marash teaches a sensor array having individual sensors which receive signals from a signal source and from interference sources (which reads on a plurality of microphones positioned to detect speech from a single speech source and noises from multiple sources) ...." Applicant concedes that Marash shows an array of sensors, as described only briefly in column 4, lines 50-67. One has to read beyond this portion of the specification to discover that the main channel matrix unit 3 and the reference channel matrix unit 4 are beam forming networks designed to select, respectively, the signal of interest from an "on-axis" signal source, and interfering signals located "off-axis." In general, beam forming requires knowledge of the array geometry, which is an aspect of the system not discussed in Marash. Thus, the sensors or microphones cannot be placed randomly if beam forming is to be used.

The Examiner next notes that Marash is characterized by "producing a main channel representing signals received in the direction of the source, such that the main channel contains both a source signal component and interference signal component (which reads on generating microphone output signals, such that the reference microphone receives acoustic signals both from the speech source and from multiple noise sources)." Yes, of course, the sensors in Marash receive signals from a source of interest and signals from interfering sources. However, the Examiner's comment also refers to "the main channel representing signals received in the direction of the source." The present invention has no such "main channel" and does not use directionality of the signal (beam steering).

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The Examiner further notes that the Marash system produces "a reference channel representing signals from directions other than that of the signal source." This has no bearing on the present invention, which does not have a reference channel representing signals from directions other than that of the signal source. The "reference" channel in Marash contains only signals received from other than the signal source direction, and obtains these signals by beam steering the sensor array to present a null in the direction of the signal source. The present invention does not perform this function.

The Examiner goes on to concede that Marash does not teach a plurality of bandpass filters for eliminating spectral bands containing noise from the microphone output signals. Applicant concedes that Coker et al. contains a statement referring to the use of filters to eliminate noise. However, in view of the serious deficiencies of the principal reference, adding Coker et al. to the cited art does not render the present claims obvious.

The Examiner further cites column 8, lines 23-67, for the contention that "Marash teaches a plurality of adaptive filters to process the output signals from the reference channels with the output signals from the main channel, which reads on the plurality of adaptive filters, for aligning each data microphone output signal with the output signal from the reference microphone." The cited text refers to an inherent time delay in processing the "reference" channels and the consequent need to delay the "main" channel to ensure that corresponding signals arrive in synchronism at the difference unit 108. "Reference" channels are those processing the interference signals in Marash. In the present invention, the reference microphone is so designated only to

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allow synchronism of the multiple microphone signals. Therefore, the only relevance of the cited material is that it refers to a time synchronism problem, but a different one.

The Examiner further cites column 6, lines 1-27, and states that "Marash teaches the processing of the main channel as a weighted sum of outputs which filters a signal coming in all directions to produce a signals coming in a specific direction, which reads on the signal summation circuit for combining filtered output signals from the microphones whereby signal components resulting from speech are combined." Applicant respectfully disagrees with this characterization of the processing performed in the "main channel" of Marash. The main channel matrix is a beam steering device that steers the sensor array to receive only signals from the direction of the signals source. How this disclosure "reads on" the signal summation circuit of the present invention is not adequately explained, perhaps because it defies explanation. The signal summation circuit combines filtered output signals from multiple microphones, but these output signals are derived from both signal and noise sources, and there is no attempt to steer the sensor array.

The Examiner further states that Marash teaches at column 6, lines 28-48, "the processing of the reference channel matrix to produce a reference channel as a weighted sum of interference signals, which reads on a signal summation circuit for combining filtered output signals from the microphones whereby signal components resulting from noise are combined." The cited text describes how the reference channel matrix unit 4 operates to produce a beam null in the direction of the beam. The signal summation circuit combines filtered output signals from multiple microphones,

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but these output signals are derived from both signal and noise sources, and there is no attempt to steer the sensor array.

In paragraph 5 of the Office action, the Examiner contends that, with respect to claims 2 and 7, in spite of the lack of any teaching of speech detection and enabling the adaptation process only when speech is detected, it would have been obvious to provide speech detection for this purpose. Applicant respectfully disagrees and also maintains that claims 2 and 7 should be allowable with the claims from which they depend.

In paragraph 6 of the Office action, the Examiner contends that, with respect to claims 3 and 8, Marash discloses (at column 8, lines 46-48) "a difference unit for subtracting the reference channel matrix (interference channel) from the main channel matrix (the source and interference channel) to effectively reduce the interference in the signal of interest, which reads on the speech conditioning circuitry coupled to the signal summation circuit to reduce reverberation effects." If the difference unit 8 is the "speech conditioning circuitry" recited in these claims, it is difficult to conceive how Marash disclosure can perform the fundamental function of reducing the effects of intereference without the difference unit. Speech conditioning circuitry is recited in claims 3 and 8 as an optional feature (not included in the corresponding independent claims). If the difference unit is needed to meet claims 1 and 6, as the Examiner contends, it surely cannot also be relied on to meet dependent claims 3 and 8. The speech conditioning circuitry has functions other than simply reducing reverberation effects and these have been added to claims 3 and 8 by way of further clarification.

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In paragraph 7 of the Office action, regarding claims 4-5 and 9-10, the Examiner contends that Marash (at column 8, line 23, through column 9, line 40) discloses a frequency selective constraint adaptive filter which implements a Least Mean Square (LMS) algorithm for adaptive filtering via updating or adjusting the filter weights. Applicant concedes only that Marash discloses the use of finite impulse response (FIR) filters, but in a totally different context from the one in which they are used in the present invention. Marash uses FIR filters to produce representations of interference signals, which are then subtracted from the "main" signal in the difference unit 8. In the present invention, adaptive (FIR) filters are used to align the data microphone output signals with the output signal from the reference microphone. This is basically a synchronization function that permits the multiple microphone signals to be combined by summation. Applicant does not maintain that adaptive FIR filters are novel in themselves, but only in the content of the claimed combination. The Examiner has cited the use of FIR filters in the context of the Marash disclosure, which, as discussed above, is not in any way suggestive of the present invention.

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CONCLUSION:

Applicant has shown that the new grounds of rejection are unfounded, and respectfully requests entry of this amendment to place the application in condition for allowance. Altematively, if the Examiner maintains the final rejection, entry of the amendment is sought to place the application in better condition for appeal. Reconsideration of the rejection in light of these remarks is respectfully solicited.

Respectfully submitted,

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ATTACHMENT FOR CLAIM AMENDMENTS
VERSION WITH MARKINGS TO SHOW CHANGES MADE
U.S. Serial No. 09/388,010; Filed: September 1, 1999

1. (Twice Amended) A microphone array processing system for performance enhancement in noisy environments, the system comprising:

a plurality of microphones positioned to detect speech from a single speech source and noise from multiple sources, and to generate corresponding microphone output signals, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone receives acoustic signals both from the speech source and from the multiple noise sources;

a plurality of bandpass filters, one for each microphone, for eliminating from the microphone output signals a known spectral band containing noise;

a plurality of adaptive filters, one for each of the data microphones, for aligning each data microphone output signal with the output signal from the reference microphone; and

a signal summation circuit, for combining the filtered output signals from the microphones, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio <u>without the need for beam steering or noise estimation techniques</u>.

3. (Amended) A system as defined in claim 1, and further comprising speech conditioning circuitry coupled to the signal summation circuit, to reduce reverberation

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effects in the output signal by modifying the spectrum of the cumulative signal obtained from the signal summation circuit.

6. (Twice Amended) A method for improving detection of speech signals in noisy environments, the method comprising:

positioning a plurality of microphones to detect speech from a single speech source and noise from multiple sources, one of the microphones being designated a reference microphone and the others being designated data microphones, wherein the reference microphone receives acoustic signals both from the speech source and from the multiple noise sources;

generating microphone output signals in the microphones;

filtering the microphone output signals in a plurality of bandpass filters, one for each microphone, to eliminate from the microphone output signals a known spectral band containing noise;

adaptively filtering the microphone output signals in a plurality of adaptive filters, one for each of the data microphones, and thereby aligning each data microphone output signal with the output signal from the reference microphone; and

combining the adaptively filtered output signals from the microphones in a signal summation circuit, whereby signal components resulting from the speech source combine coherently and signal components resulting from noise combine incoherently, to produce an increased signal-to-noise ratio without the need for beam steering or noise estimation techniques.

8. (Amended) A method as defined in claim 6, and further comprising the step of conditioning the combined signals in speech conditioning circuitry coupled to the

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signal summation circuit, to reduce reverberation effects in the output signal by modifying the spectrum of the cumulative signal obtained from the signal summation circuit.

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